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FROM:Whitham, Curtis & Christofferson, P.C.

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Docket No. SON-0432 In re: ☒ patent/ ☐ trademark application ofApplicant(s) OzawaSerial No. 09/302397 Date Filed 4-30-00Papers filed herewith on 12-3-01

☒ Fees \$ Deposit Account No. (if applicable) 50-2041
(filing fee; Assignment charge; 110 Extension of Time;
 issue fee/advance copies; other)

☒ Amendment ☐ Notice of Appeal ☐ Appeal Brief
☐ Sheets of Drawings ☐ Proposed Drawing Corrections (w/ drawings)
☒ Change of Address ☒ Request for Extension of Time 1-month
☐ Assignment ☐ Recordation Form Cover Sheet
☐ Information Disclosure Statement ☐ PTO-1449 and associated art (docs.)
☐ Priority Document(s) ☐ Other

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T1D-115496M/SKY

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re patent application of

OZAWA, K.

Serial No. 09/302,397

Filed: April 30, 1990

For: SPEECH CODING APPARATUS AND SPEECH DECODING
APPARATUS

Examiner: Armstrong, A.

Group Art Unit: 2641

Assistant Commissioner of Patents
Washington, D.C. 20231

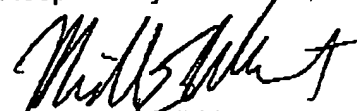
PETITION AND FEE FOR EXTENSION OF TIME

Sir:

In response to the Office Action mailed July 3, 2001, an Amendment is being submitted concurrently herewith. Applicant hereby petitions for a one-month extension of time to respond.

Applicants hereby make a written conditional petition for any additional extension of time deemed necessary. Please charge the Extension of Time fee and any deficiencies in fees and/or credit any overpayment of fees to attorney's Deposit Account 50-2041.

Respectfully submitted,



Michael E. Whitham

Registration No.: 32,635

WHITHAM, CURTIS &
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11491 Sunset Hills Road, Suite 340
Reston, Virginia 20190

703-787-9400

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OZAWA, K. :
Serial No. 09/302,397 : Examiner: Armstrong, A.
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NOTICE OF CHANGE OF FIRM NAME AND ADDRESS

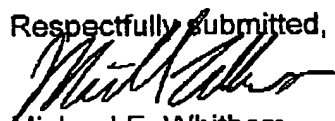
Sir:

Please be advised that the firm name and correspondence address of attorneys of record in the above-identified application has been changed to:

WHITHAM, CURTIS & CHRISTOFFERSON, P.C.
11491 Sunset Hills Road, Suite 340
Reston, Virginia 20190

Additionally, please remove Joseph M. Martinez de Andino, Registration No. 37,178 as attorney of record.

Respectfully submitted,


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IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re patent application of :
OZAWA, K.)
Serial No. 09/302,397 : Examiner: Armstrong, A.
Filed: April 30, 1990 : Group Art Unit: 2641
For: SPEECH CODING APPARATUS AND SPEECH DECODING APPARATUS

AMENDMENT

Assistant Commissioner for Patents
Washington, D.C. 20231

Sir:

The following amendments and remarks are submitted in response to the Office Action
mailed on July 3, 2001 in connection with the above-identified application.

IN THE CLAIMS

Please amend claims 1-7 as follows. A clean version of the amended claims is attached
to this paper.

1. (Amended) A speech coding apparatus including at least:
a spectrum parameter calculation section for receiving a speech signal, obtaining
a spectrum parameter, and quantizing the spectrum parameter,
an adaptive codebook section for obtaining a delay and a gain from a past
quantized sound source signal by using an adaptive codebook, and obtaining a residue by
predicting a speech signal, and

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a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal, comprising:

a discrimination section for discriminating a voiced sound mode and an unvoiced sound mode on a [the] basis of a past quantized gain of an adaptive codebook;

a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on [when] an output from said discrimination circuit section [indicates a predetermined mode], and searches combinations of code vectors stored in said codebook and a plurality of shift amounts used to shift positions of the pulses so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech; and

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section.

2. (Amended) A speech-coding apparatus including at least:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal, and

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a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal, comprising:

a discrimination section for discriminating a voice sound mode and an unvoiced sound mode on a [the] basis of a past quantized gain of an adaptive codebook;

a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on [when] an output from said discrimination section [indicates a predetermined mode], and outputs a code vector that minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule; and

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section

3. (Amended) A speech coding apparatus including at least:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal, and

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a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal, comprising:

a discrimination section for discriminating a voice sound mode and an unvoiced sound mode on the basis of a past quantized gain of an adaptive codebook;

a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on [when] an output from said discrimination section [indicates a predetermined mode], and a gain codebook for quantizing gains, and searches combinations of code vectors stored in said codebook, a plurality of shift amounts used to shift positions of the pulses, and gain code vectors stored in said gain codebook so as to output a combination of a code vector, shift amount, and gain code vector which minimizes distortion relative to input speech; and

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section.

4. (Amended) A speech coding apparatus including at least;

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

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an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal, and

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal, comprising:

a discrimination section for discriminating a voice sound mode and an unvoiced sound mode on the basis of a past quantized gain of a adaptive codebook;

a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on [when an] output from said discrimination section [indicates a predetermined mode], and a gain codebook for quantizing gains, and outputs a combination of a code vector and gain code vector which minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule; and

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section.

5. (Amended) A speech decoding apparatus comprising:

a demultiplexer section for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information;

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a mode discrimination section for discriminating a voice sound mode and an unvoiced sound mode by using a past quantized gain in said adaptive codebook; and

a sound source signal reconstructing section for reconstructing a sound source signal by generating non-zero pulses from the quantized sound source information based on [when] an output from said discrimination section [indicates a predetermined mode],

wherein a speech signal is reproduced by passing the sound source signal through a synthesis filter section which includes [constituted by] spectrum parameters.

6. (Amended) A speech coding/decoding apparatus comprising:

a speech coding apparatus including:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal,

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal,

a discrimination section for discriminating a voice sound mode and an unvoiced sound mode on the basis of a past quantized gain of a adaptive codebook, and

a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses when an output from said discrimination section indicates a predetermined mode,

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said sound source quantization section searching combinations of code vectors stored in said codebook and a plurality of shift amounts used to shift positions of the pulses so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech, and further including:

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section; and

a speech decoding apparatus including at least:

a demultiplexer section for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information,

a mode discrimination section for discriminating a mode by using a past quantized gain in said adaptive codebook,

a sound source signal reconstructing section for reconstructing a sound source signal by generating non-zero pulses from the quantized sound source information when an output from said discrimination indicates a predetermined mode, and

a synthesis filter section which is constituted by spectrum parameters and reproduces a speech signal by filtering the sound source signal.

7. (Amended) A speech coding/decoding apparatus comprising:

a speech coding apparatus including:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

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an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal,

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal,

a discrimination section for discriminating a voice sound mode and an unvoiced sound mode on the basis of a past quantized gain of a adaptive codebook, and

a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on [when] an output from said discrimination section [indicates a predetermined mode],

said sound source quantization section for outputting a combination of a code vector and shift amount which minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule, and further including

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section; and

a speech decoding apparatus including at least:

a demultiplexer section for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information,

a mode discrimination section for discriminating a mode by using a past quantized gain in said adaptive codebook,

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a sound source signal reconstructing section for reconstructing a sound source signal by generating positions of pulses according to a predetermined rule and generating amplitudes or polarities for the pulses from a code vector when an output from said discrimination section indicates a predetermined mode, and

a synthesis filter section which includes [is constituted by] spectrum parameters and reproduces a speech signal by filtering the sound source signal.

REMARKS

Claims 1-11 are pending, of which claims 1-7 have been amended.

Reconsideration of the application is respectfully requested for the following reasons.

In the Office Action, the Examiner rejected claims 1-11 under 35 USC § 103(a) for being obvious in view of a combination formed between the Kleijn and Swaminathan patents. Applicant traverses this rejection for the following reasons.

As disclosed in the Background of the Invention section of the specification, conventional coding methods have drawbacks relating to the manner in which background noise is processed. Specifically, these methods represent a sound source signal as a plurality of pulses which concentrate near a pitch pulse. This pitch pulse, in turn, serves as the start point of a pitch in a vowel interval of speech. When the sound source signal corresponds to input speech, these methods reduce the number of processing calculations required. However, when the sound source signal is background noise (or some other randomly generated signal), conventional

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methods cannot express this noise by a small number of pulses. As a result, if the bit rate decreases, and the number of pulses decreases, the sound quality of background noise abruptly deteriorates. (See, for example, page 4, lines 10-21).

Applicant's invention overcomes these drawbacks by preventing sound quality from deteriorating when background noise is processed. In accordance with one aspect of the invention, sound quality is kept at a high level (higher than that attainable by conventional methods) by requiring a relatively few number of calculations to be performed when processing background noise. This is accomplished at least through the mode discrimination section and the sound source quantization section of the invention. These features which enable the invention to outperform conventional methods are recited in the claims.

Claim 1 recites a speech coding apparatus including at least a spectrum parameter calculation section, an adaptive codebook section, and a sound source quantization section. The spectrum parameter calculation section receives a speech signal, obtains a spectrum parameter, and quantizes the spectrum parameter. The adaptive codebook section obtains a delay and a gain from a past quantized sound source signal by using an adaptive codebook and obtains a residue by predicting a speech signal. The sound source quantization section quantizes a sound source signal of the speech signal by using the spectrum parameter and outputs the sound source signal.

Claim 1 further recites that the sound source quantization section includes a discrimination section, a sound source quantization section, and a multiplexer section. The discrimination section discriminates a voiced sound mode and an unvoiced sound mode on a basis of a past quantized gain of an adaptive codebook. The sound source quantization section has a codebook for representing a sound source signal by a combination of a plurality of non-

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zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination circuit section, and searches combinations of code vectors stored in said codebook and a plurality of shift amounts used to shift positions of the pulses so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech. The multiplexer section outputs a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section.

To render claim 1 obvious, two requirements must be satisfied. First, the cited references must teach or suggest all the features in claim 1. Second, there must have been some teaching or suggestion in existence at the time the claimed invention was made that would have led one of ordinary skill in the art to combine the references in an attempt to form the invention. See MPEP § 2143.01 and *In re Rouffet*, 47 USPQ.2d 1459 (Fed. Cir. 1997).

The Kleijn patent discloses a method for coding speech signals which provides a peak-to-average ratio criterion that determines whether or not time shifting of a speech residual signal should be applied within a certain sub-frame.

Claim 1 is different from Kleijn in at least four respects.

First, claim 1 recites a discrimination section for discriminating a voiced sound mode and an unvoiced sound mode. This discrimination function is performed so that, for example, different processing functions may be employed to minimize distortion during speech coding. More specifically, the invention processes an input speech signal differently (e.g., uses different pulse-shifting schemes) based on whether a voiced sound mode is detected or whether an unvoiced sound mode is detected. (This unvoiced sound mode may, for example, correspond

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to a situation where the input speech signal contains only random noise, e.g., background noise.) Consistent with the advantages of invention discussed above, by detecting a voiced sound mode and an unvoiced sound mode relative to an input signal, the claimed invention may process the signal differently in order to attain optimum signal quality preservation.

The Kleijn patent does not teach or suggest the discrimination section of the claimed invention. In the Office Action, the Examiner drew a correlation between this discrimination section and the features of the Kleijn system discussed at column 7, lines 10-26. This portion of Kleijn discloses that a value G_{opt} is calculated based on a linear interpolation of linear pitch values at or near each frame boundary. This G_{opt} value is then compared with a threshold value, but not for purposes of discriminating between a voiced sound mode or unvoiced sound mode as recited in claim 1. Instead, this comparison is made in order to calculate a peak-to-average ratio of a residual sound signal $R(n)$, which is completely unrelated to determining whether an input speech signal was received in a voiced sound or unvoiced sound mode. Thus, it is clear that neither column 7 nor any other portion of the Kleijn patent teaches or suggest the discrimination section of claim 1.

Second, claim 1 recites that the discrimination section discriminates a voiced sound mode and an unvoiced sound mode based on a past quantized gain of an adaptive codebook. The Kleijn patent does not teach or suggest performing the function of discriminating between voiced sound and unvoiced sound modes. It is therefore logically follows that Kleijn does not teach or suggest perform this function based on past quantized gain of an adaptive codebook.

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Third, claim 1 recites a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination section. In determining an optimum value for a matching criterion for purposes of time-shifting pulses, the Kleijn method performs the function of adjusting the amplitude of an adaptive codebook vector $e(n)$. (See column 6, lines 54-56, which was, in part, relied on by the Examiner to support the position that Kleijn discloses the sound source quantization section of the claimed invention.) Unlike the claimed invention, however, Kleijn does not have the discrimination section of the claimed invention and thus does not teach or suggest collectively quantizing amplitudes or polarities of pulses based on an output of such a discrimination section.

Fourth, claim 1 further recites that the sound source quantization section searches combinations of code vectors stored in said codebook and a plurality of shift amounts used to shift positions of the pulses so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech. The Kleijn patent also fails to teach or suggest these features.

As previously discussed, the claimed invention processes pulses differently, for example, by using different pulse-shifting schemes depending upon whether a voice sound mode or an unvoiced sound mode is discriminated. Either way, a time-shifting scheme is employed. In Kleijn, however, two modes of operation were identified by the Examiner as a result of the reference to column 7. The first mode is where $G_{opt} < \text{a threshold}$ and the second mode is where $G_{opt} < \text{a threshold}$. Contrary to the claimed invention, Kleijn discloses that no time shift will be performed when G_{opt} is found to be smaller than the threshold. So, in at least one of its modes

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of operation (e.g., $G_{opt} < \text{threshold}$), no time shift will occur. Accordingly, it is clear that Kleijn does not teach or suggest performing the claimed function of searching "a plurality of shift amounts used to shift positions of the pulses so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech" when operating in both of its modes of operation.

Based on at least the foregoing differences, it is respectfully submitted that claim 1 is patentably distinguishable from the Kleijn patent. To make up for the deficiencies of Kleijn, the Swaminathan patent was cited.

The Swaminathan patent was cited for its disclosure of a multiplexer and decoding scheme. This patent, however, does not teach or suggest the four features of claim 1 that are missing from the Kleijn patent. It is therefore respectfully submitted that any combination formed between the Kleijn and Swaminathan patents would also fail to include these features. For these reasons, it is respectfully submitted that claim 1 is allowable over a Kleijn-Swaminathan combination.

Claim 2 recites the first three features discussed above which render claim 1 patentably distinguishable from a Kleijn-Swaminathan combination. In addition to these features, claim 2 recites a sound source quantization section which outputs a code vector that minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule. (These features are disclosed, for example, at page 26, lines 5-12, of Applicant's specification, where it is disclosed that pulse positions are set at *predetermined intervals* during an unvoiced sound mode, and that shift amounts for shifting positions of the pulses are determined *in advance*.) (Emphasis added.)

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The Kleijn patent does not teach or suggest these features. The Kleijn method performs time shifts only when $G_{opt} > \text{threshold}$. When this occurs, Kleijn discloses that time shifts are performed based on a comparison of a plurality of temporally-distorted versions of residual signal $r(n)$ and delayed residual signal $R(n - D(n))$. A matching criterion is then determined, and then the threshold comparison is made with respect to G_{opt} to determine whether a time-shift operation should be performed. From these disclosures, it is therefore clear that when Kleijn does perform a time-shift operation, the time shift is not performed based on a predetermined rule as recited in claim 2, i.e., based on time shift positions determined in advance. (The predetermined time shift positions are illustratively shown by the values in Table 2 on page 26 of Applicant's specification).

The Swaminathan patent also fails to teach or suggest these features.

For at least the foregoing reasons, it is respectfully submitted that claim 2 is patentably distinguishable from a Kleijn-Swaminathan combination.

Claims 3-8 and the dependent claims recite features similar to those which render claim 1 patentably distinguishable from a Kleijn-Swaminathan combination. For at least these reasons, it is respectfully submitted that claims 3-11 allowable over the cited combination.

Reconsideration and withdrawal of all the rejections and objections made by the Examiner is hereby respectfully requested.

In view of the foregoing amendments and remarks, it is respectfully submitted that the application is in condition for allowance. Favorable consideration and prompt allowance of the application is respectfully requested.

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Should the Examiner believe that further amendments are necessary to place the application in condition for allowance, or if the Examiner believes that a personal interview would be advantageous in order to more expeditiously resolve any remaining issues, the Examiner is invited to contact Applicants' undersigned attorney at the telephone number listed below.

To the extent necessary, Applicants petition for an extension of time under 37 CFR § 1.136. Please charge any shortage in fees due in connection with this application, including extension of time fees, to Deposit Account No. 50-2041 (Attorney Docket No. 02100033AA) and credit any excess fees to the same Deposit Account.

Respectfully submitted,



Michael E. Whitham
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Clean Version of the Amended Claims

1. (Amended) A speech coding apparatus including at least:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal, and

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal, comprising:

a discrimination section for discriminating a voiced sound mode and an unvoiced sound mode on a basis of a past quantized gain of an adaptive codebook;

a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination circuit section, and searches combinations of code vectors stored in said codebook and a plurality of shift amounts used to shift positions of the pulses so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech; and

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a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section.

2. (Amended) A speech-coding apparatus including at least:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal, and

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal, comprising:

a discrimination section for discriminating a voice sound mode and an unvoiced sound mode on a basis of a past quantized gain of an adaptive codebook;

a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination section, and outputs a code vector that minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule; and

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a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section.

4. (Amended) A speech coding apparatus including at least:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal, and

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal, comprising:

a discrimination section for discriminating a voice sound mode and an unvoiced sound mode on the basis of a past quantized gain of a adaptive codebook;

a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on output from said discrimination section, and a gain codebook for quantizing gains, and outputs a combination of a code vector and gain code vector which minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule; and

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a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section.

5. (Amended) A speech decoding apparatus comprising:

a demultiplexer section for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information;

a mode discrimination section for discriminating a voice sound mode and an unvoiced sound mode by using a past quantized gain in said adaptive codebook; and

a sound source signal reconstructing section for reconstructing a sound source signal by generating non-zero pulses from the quantized sound source information based on an output from said discrimination section,

wherein a speech signal is reproduced by passing the sound source signal through a synthesis filter section which includes spectrum parameters.

6. (Amended) A speech coding/decoding apparatus comprising:

a speech coding apparatus including:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal,

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a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section; and

a speech decoding apparatus including at least:

a demultiplexer section for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information,

a mode discrimination section for discriminating a mode by using a past quantized gain in said adaptive codebook,

a sound source signal reconstructing section for reconstructing a sound source signal by generating positions of pulses according to a predetermined rule and generating amplitudes or polarities for the pulses from a code vector when an output from said discrimination section indicates a predetermined mode, and

a synthesis filter section which includes spectrum parameters and reproduces a speech signal by filtering the sound source signal.